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Edler 1-4

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Patent Application

Applicant(s): Edler et al.
Case: 1-4
Serial No.: 09/586,072
Filing Date: June 2, 2000
Group: 2654
Examiner: Qi Han

I hereby certify that this paper is being deposited on this date with the U.S. Postal Service as first class mail addressed to the Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450

Signature: Dina Maurio Date: September 15, 2005

Title: Perceptual Coding of Audio Signals Using Separated Irrelevancy Reduction and Redundancy Reduction

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TRANSMITTAL OF APPEAL BRIEF

Mail Stop Appeal Brief
Commissioner of Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Sir:

Submitted herewith are the following documents relating to the above-identified patent application:

1. Appeal Brief (original and two copies);
2. Copy of Notice of Appeal, filed on July 13, 2005, with copy of stamped return postcard indicating receipt of Notice by PTO on July 18, 2005.

There is an additional fee of $500 due in conjunction with this submission under 37 CFR §1.17(c). Please charge **Deposit Account No. 50-0762** the amount of $500, to cover this fee. In the event of non-payment or improper payment of a required fee, the Commissioner is authorized to charge or to credit **Deposit Account No. 50-0762** as required to correct the error. A duplicate copy of this letter and two copies of the Appeal Brief are enclosed.

Respectfully submitted,

Kevin M. Mason

Date: September 15, 2005

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APPEAL BRIEF

Mail Stop Appeal Brief - Patents
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Sir:

Applicants hereby appeal the final rejection dated March 28, 2005, of claims 1 through 33 of the above-identified patent application.

REAL PARTY IN INTEREST

The present application is assigned to Agere Systems Inc., as evidenced by the statement under 37 CFR 3.73 (b) submitted on July 2, 2003. The assignee, Agere Systems Inc., is the real party in interest.

RELATED APPEALS AND INTERFERENCES

There are no related appeals or interferences.

STATUS OF CLAIMS

Claims 1 through 33 are pending in the above-identified patent application. The amendment filed on July 7, 2003, under 35 U.S.C. 132 remains objected to because it

introduces new matter into the disclosure. Claim 7 remains rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement and claims 1, 13, 20, 25, and 30-33 remain rejected under 35 U.S.C. §112, first paragraph, as failing to comply with the written description requirement and under 35 U.S.C. §112, second paragraph, as failing to set forth the subject matter which Applicants regard as the invention. Claims 1-2, 6-10, 13-14 and 30-31 remain rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. (IEEE Transaction on Signal Processing, vol. 46, April, 1998), in view of Smith (ISBN 0-9660176-33, 1997) in view of Tsurushima et al. (United States Patent Application Number 2001/0047256 A1), in view of Johnston (United States Patent Number 5,481,614), claims 5, 11-12, and 17-19 remain rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. in view of Smith, Tsurushima et al., and Johnston, and further in view of admitted prior art, and claims 3-4, 15-16, 20-29, and 32-33 remain rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. in view of Smith, Tsurushima et al., and Johnston, and further in view of well known prior art.

STATUS OF AMENDMENTS

There have been no amendments filed subsequent to the final rejection.

SUMMARY OF CLAIMED SUBJECT MATTER

The present invention is directed to a perceptual audio coder for encoding audio signals, such as speech or music, with different spectral and temporal resolutions for redundancy reduction and irrelevancy reduction. The disclosed perceptual audio coder separates the psychoacoustic model (irrelevancy reduction) from the redundancy reduction, to the extent possible. The audio signal is initially spectrally shaped using a prefilter controlled by a psychoacoustic model. The prefilter output samples are thereafter quantized and coded to minimize the mean square error (MSE) across the spectrum (page 5, lines 14-29). The disclosed perceptual audio coder can use fixed quantizer step-sizes, since spectral shaping is performed by the pre-filter prior to quantization and coding (page 6, lines 1-18). The disclosed pre-filter and post-filter support the appropriate frequency dependent temporal and spectral resolution for irrelevancy reduction. A filter structure based on a frequency-warping

technique is used that allows filter design based on a non-linear frequency scale. The characteristics of the pre-filter may be adapted to the masked thresholds (as generated by the psychoacoustic model), using techniques known from speech coding, where linear-predictive coefficient (LPC) filter parameters are used to model the spectral envelope of the speech signal. Likewise, the filter coefficients may be efficiently transmitted to the decoder for use by the post-filter using well-established techniques from speech coding, such as an LSP (line spectral pairs) representation, temporal interpolation, or vector quantization (page 6, line 19, to page 8, line 27).

STATEMENT OF GROUNDS OF REJECTION TO BE REVIEWED ON APPEAL

The amendment filed on July 7, 2003, under 35 U.S.C. 132 is objected to because it introduces new matter into the disclosure. Claim 7 is rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement and claims 1, 13, 20, 25, and 30-33 are rejected under 35 U.S.C. §112, first paragraph, as failing to comply with the written description requirement and under 35 U.S.C. §112, second paragraph, as failing to set forth the subject matter which Applicants regard as the invention. Claims 1-2, 6-10, 13-14 and 30-31 are rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al., in view of Smith, Tsurushima et al., and Johnston, claims 5, 11-12, and 17-19 are rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. in view of Smith, Tsurushima et al., and Johnston, and further in view of admitted prior art, and claims 3-4, 15-16, 20-29, and 32-33 are rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. in view of Smith, Tsurushima et al., and Johnston, and further in view of well known prior art.

ARGUMENT

Formal Objections

The amendment filed on July 7, 2003, is objected to under 35 U.S.C. 132 because it introduces new matter into the disclosure. In particular, the Examiner asserts that the following statement is added material:

In the case of an image signal, the adaptive filter is controlled in a way that the magnitude response approximates an inverse of a corresponding visibility threshold, as would be apparent to a person of ordinary skill in the art. (Substitute Specification, page 7, lines 23-25.)

Appellants note that the cited statement was disclosed in claim 7 of the original disclosure, and thus does not constitute new matter. Appellants also note that standard video and image coding algorithms (e.g., JPEG and MPEG) code an image or a frame of a video signal (or a prediction error from motion compensated temporal prediction) using a two-dimensional transform (most commonly, a Discrete Cosine Transform). For adapting the spectral shape (spatial frequency) of the quantization error to the visibility threshold, a weighting matrix is applied before quantization. *See also*, the method described by Srinivasan and referred to by the examiner. Thus, it would be obvious to a person of ordinary skill in the art to replace the one-dimensional pre-filter applied before the one-dimensional transform in audio coding by a two-dimensional spatial pre-filter before the two-dimensional transform in video coding. Appellants therefore respectfully request that the objection under 35 U.S.C. 132 be withdrawn.

<u>Section 112 Rejections</u>

Claim 7 is rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement. In particular, the Examiner asserts that an audio signal processed in one dimension is very different from an image signal processed in two dimensions, including filters, transforms, algorithms, and the related hardware and software, which is not disclosed in the specification and, thus, the Appellants' specification does not disclose the claimed subject matter in such full, clear, concise, and exact terms as to enable any person skilled in the art to make and/or use the claimed invention, without undue effort.

Claims 1, 13, 20, 25, and 30-33 are rejected under 35 U.S.C. §112, second paragraph, as failing to set forth the subject matter which applicant regards as the invention. In particular, the Examiner asserts that passages cited in the "Field of the Invention" and "Summary of Invention" sections indicate that the invention is different from what is defined in the claims because no audio coding (encoding or decoding) or audio signal is recited in

said independent claims. The Examiner further asserts that the limitation "the spectoral and temporal resolutions ..." lacks sufficient antecedent basis.

As noted above, standard video and image coding algorithms (e.g., JPEG and MPEG) code an image or a frame of a video signal (or a prediction error from motion compensated temporal prediction) using a two-dimensional transform (most commonly a Discrete Cosine Transform). For adapting the spectral shape (spatial frequency) of the quantization error to the visibility threshold, a weighting matrix is applied before quantization. This is very similar to the method described by Srinivasan and referred to by the examiner. A person of ordinary skill in the art would therefore recognize that the one-dimensional pre-filter applied before the one-dimensional transform in audio coding could be replaced by a two-dimensional spatial pre-filter before the two-dimensional transform to apply the methods of the present invention to video coding. Thus, Appellants maintain that the specification discloses the claimed subject matter in such full, clear, concise, and exact terms as to enable any person skilled in the art to make and/or use the claimed invention, without undue effort. In addition, since the methods of the present invention can be applied to both audio coding and video coding, it is not necessary to limit the claims to audio encoding/decoding.

Claims 1, 13, 20, 25, and 30-33 are also rejected under 35 U.S.C. §112, first paragraph, as failing to comply with the written description requirement. In particular, the Examiner asserts that the limitation (claim 1) "wherein the spectral and temporal resolutions of one or more subbands utilized in said encoding are selected independent of said adaptive filter" introduces new subject matter, because it is not specifically disclosed in the original specification.

In the present invention, even though subband signals are also quantized, the quantizer step sizes are not adapted, because it is desired to have white-noise-like quantization error at the output of the synthesis filter bank. Since the temporal and spectral shaping of the quantization error is performed by the post-filter, *which can be controlled without any dependence on the filter bank, the spectoral and temporal resolutions utilized in the encoding are independent of the adaptive filter, as would be apparent to a person of ordinary skill in the art*. Thus, the limitation "wherein the spectral and temporal resolutions

of one or more subbands utilized in said encoding are selected independent of said adaptive filter" is not new subject matter.

Regarding the cited limitation that lacks sufficient antecedent basis, Appellants propose to correct the antecedent basis upon resolution of the appeal. Appellants therefore respectfully request that the rejections under section 112 be withdrawn.

Independent Claims 1, 13, 20, 25 and 30-33

Independent claims 1, 13, and 30-31 are rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al., in view of Smith in view of Tsurushima et al., in view of Johnston, and claims 20, 25, and 32-33 are rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. in view of Smith in view of Tsurushima et al. in view of Johnston, and further in view of well known prior art.

Regarding claim 1, the Examiner asserts that Srinivasan et al. teach an adaptive filter producing a filter output signal and having a magnitude response. The Examiner acknowledges that Srinivasan in view of Smith does not expressly disclose "said inverse of said certain form of a signal being or relating an inverse of 'the masking threshold' and 'the shapes of the masking thresholds are almost frequency independent,'" but asserts that this feature is well known in the art as evidenced by Tsurushima who discloses...(which suggests that the shapes of the masking thresholds are ***almost*** *frequency independent*) (emphasis added).

Appellants note that, in the method taught by Srinivasan, the spectral and temporal resolutions used for coding can not be selected *independent* of the adaptive filter, *since the same subbands are used for coding and for noise shaping*. Independent claims 1, 13, 20, 25 and 30-33 require wherein the spectral and temporal resolutions of one or more subbands utilized in said encoding (decoding) are selected ***independent*** of said adaptive filter. As the Examiner acknowledges, the cited prior art does not teach that the spectral and temporal resolutions of one or more subbands utilized in said encoding (decoding) are selected ***independent*** of said adaptive filter. Appellants also note that, in the system taught by Srinivasan, there is no signal corresponding to the pre-filter output and that *only the output* of the overall system has similarly shaped noise.

Appellants also note that, in every case, where subband signals are quantized with quantizer step sizes adapted to masked thresholds, this adaptation is limited by the spectral and temporal resolution of the synthesis filter bank. This is due to the fact that the quantized/reconstructed signal in the decoder has to go through the synthesis filter bank.

For example, in an (assumed) extreme case, it is desirable in a certain time interval to have a white-noise-like quantization error (i.e. constant over time and frequency). If it is now desirable to have (e.g. due to an indication by a psychoacoustic model) a significantly larger error for one point in time and at a specific frequency, the quantizer step size would be increased only for one single subband sample in the corresponding subband. This would lead to an impulse-like quantization error in this subband. Therefore, the resulting quantization error in the filter bank output, which would superimpose the white-noise-like error, would be the impulse response of the corresponding synthesis filter, weighted with the magnitude of the single large quantization error. Therefore, its temporal and spectral distribution is equivalent to that of the synthesis filter impulse response. A higher resolution cannot be obtained.

In the present invention, even though subband signals are also quantized, the quantizer step sizes are not adapted, because it is desired to have white-noise-like quantization error at the output of the synthesis filter bank. The temporal and spectral shaping of the quantization error is performed by the post-filter, *which can be controlled without any dependence on the filter bank.* This independence, as required by the independent claims of the present invention, is not disclosed or suggested by Srinivasan et al., Smith, Tsurushima et al., Johnston, and the cited well known prior art (alone or in combination).

Thus, Srinivasan et al., Smith, Tsurushima et al., Johnston, and the cited well known prior art (alone or in combination) do not disclose or suggest wherein the spectral and temporal resolutions of one or more subbands utilized in said encoding (decoding) are selected independent of said adaptive filter, as required by independent claims 1, 13, 20, 25 and 30-33.

Additional Cited References

Smith was also cited by the Examiner for its disclosure of custom filters...comprising a deconvolution filter having a frequency response which has an inverse response part. Appellants note that Smith does not address the issue of utilizing spectral and temporal resolutions of one or more subbands that are selected independent of an adaptive filter for encoding or decoding.

Thus, Smith does not disclose or suggest wherein the spectral and temporal resolutions of one or more subbands utilized in said encoding (decoding) are selected independent of said adaptive filter, as required by independent claims 1, 13, 20, 25 and 30-33.

Tsurushima was also cited by the Examiner for its disclosure of "a combination of a convolution filter circuit 523, divider 526 for deconvolving the allowable noise level, and subtractor 528 subtracts the masking threshold from the Bark spectrum SB for masking the portions of the spectral components SB lower than the level of the masking spectrum MS." Appellants note that Tsurushima does not address the issue of utilizing spectral and temporal resolutions of one or more subbands that are selected independent of an adaptive filter for encoding or decoding.

Thus, Tsurushima does not disclose or suggest wherein the spectral and temporal resolutions of one or more subbands utilized in said encoding (decoding) are selected independent of said adaptive filter, as required by independent claims 1, 13, 20, 25 and 30-33.

Johnston was also cited by the Examiner for its disclosure of a method and apparatus for coding audio signals based on a perceptual model. Appellants note that Johnston does not address the issue of utilizing spectral and temporal resolutions of one or more subbands that are selected independent of an adaptive filter for encoding or decoding.

Thus, Johnston does not disclose or suggest wherein the spectral and temporal resolutions of one or more subbands utilized in said encoding (decoding) are selected independent of said adaptive filter, as required by independent claims 1, 13, 20, 25 and 30-33.

Conclusion

Thus, Srinivasan et al., Smith, Tsurushima et al., Johnston, and well known prior art, alone or in combination, do not disclose or suggest wherein the spectral and temporal resolutions of one or more subbands utilized in said encoding (decoding) are selected independent of said adaptive filter, as required by independent claims 1, 13, 20, 25 and 30-33.

The rejections of the independent claims under section §103 in view of Srinivasan et al., Smith, Tsurushima et al., Johnston, and well known prior art, alone or in any combination, are therefore believed to be improper and should be withdrawn. The remaining rejected dependent claims are believed allowable for at least the reasons identified above with respect to the independent claims.

The attention of the Examiner and the Appeal Board to this matter is appreciated.

Respectfully,

Kevin M. Mason

Date: September 15, 2005

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APPENDIX

1. A method for encoding a signal, comprising the steps of:

filtering said signal using an adaptive filter controlled by a psychoacoustic model, said adaptive filter producing a filter output signal and having a magnitude response that approximates an inverse of the masking threshold; and

quantizing and encoding the filter output signal together with side information for filter adaptation control, wherein the spectral and temporal resolutions of one or more subbands utilized in said encoding are selected independent of said adaptive filter.

2. The method of claim 1, wherein said quantizing and encoding step uses a transform or analysis filter bank suitable for redundancy reduction.

3. The method of claim 1, further comprising the steps of quantizing and encoding spectral components obtained from a transform or analysis filter bank, and wherein said quantizing and encoding steps employ fixed quantizer step sizes.

4. The method of claim 1, wherein said quantizing and encoding step reduces the mean square error in said signal.

5. The method of claim 1, wherein a filter order and intervals of filter adaptation of said adaptive filter are selected suitable for irrelevancy reduction.

6. The method of claim 1, wherein said signal is an audio signal.

7. The method of claim 1, wherein said signal is an image signal and said adaptive filter is controlled in a way that said magnitude response approximates an inverse of a visibility threshold.

8. The method of claim 1, further comprising the step of transmitting said encoded signal to a decoder.

9. The method of claim 1, further comprising the step of recording said encoded signal on a storage medium.

10. The method of claim 1, wherein said encoding further comprises the step of employing an adaptive Huffman coding technique.

11. The method of claim 1, wherein said filtering step is based on a frequency-warping technique using a non-linear frequency scale.

12. The method of claim 1, wherein the encoding stage for filter coefficients comprises a conversion from linear-predictive coefficient filter coefficients to lattice coefficients or to Line Spectrum Pairs.

13. A method for encoding a signal, comprising the steps of:

filtering said signal using an adaptive filter controlled by a psychoacoustic model, said adaptive filter producing a filter output signal and having a magnitude response that approximates an inverse of the masking threshold; and

transforming the filter output signal using a plurality of subbands suitable for redundancy reduction; and

quantizing and encoding the subband signals together with side information for filter adaptation control, wherein the spectral and temporal resolutions of one or more subbands utilized in said encoding are selected independent of said adaptive filter.

14. The method of claim 13, wherein said quantizing and encoding step uses a transform or analysis filter bank suitable for redundancy reduction.

15. The method of claim 13, further comprising the steps of quantizing and encoding spectral components obtained from a transform or analysis filter bank, and wherein said quantizing and encoding steps employ fixed quantizer step sizes.

16. The method of claim 13, wherein said quantizing and encoding step reduces the mean square error in said signal.

17. The method of claim 13, wherein a filter order and intervals of filter adaptation of said adaptive filter are selected suitable for irrelevancy reduction.

18. The method of claim 13, wherein said filtering step is based on a frequency-warping technique using a non-linear frequency scale.

19. The method of claim 13, wherein the encoding stage for filter coefficients comprises a conversion from linear-predictive coefficient filter coefficients to lattice coefficients or to Line Spectrum Pairs.

20. A method for decoding a signal, comprising the steps of:

decoding and dequantizing said signal;

decoding side information for filter adaptation control transmitted with said signal; and

filtering the dequantized signal with an adaptive filter controlled by said decoded side information, said adaptive filter producing a filter output signal and having a magnitude response that approximates the masking threshold, wherein the spectral and temporal resolutions of one or more subbands utilized in said decoding are selected independent of said adaptive filter.

21. The method of claim 20, wherein said decoding and dequantizing step uses an inverse transform or synthesis filter bank suitable for redundancy reduction.

22. The method of claim 20, further comprising the steps of decoding and dequantizing spectral components obtained from a transform or synthesis filter bank, and wherein said decoding and dequantizing steps employ fixed quantizer step sizes.

23. The method of claim 20, wherein a filter order and intervals of filter adaptation of said adaptive filter are selected suitable for irrelevancy reduction.

24. The method of claim 20, wherein the decoding stage for filter coefficients comprises a conversion from lattice coefficients or to Line Spectrum Pairs to linear-predictive coefficient filter coefficients.

25. A method for decoding a signal transmitted using a plurality of subband signals, comprising the steps of:

decoding and dequantizing said transmitted subband signals;

decoding side information for filter adaptation control transmitted with said signal;

transforming said subbands to a filter input signal; and

filtering the filter input signal with an adaptive filter controlled by said decoded side information, said adaptive filter producing a filter output signal and having a magnitude response that approximates the masking threshold, wherein the spectral and temporal resolutions of one or more subbands utilized in said decoding are selected independent of said adaptive filter.

26. The method of claim 25, wherein said decoding and dequantizing step uses an inverse transform or synthesis filter bank suitable for redundancy reduction.

27. The method of claim 25, further comprising the steps of decoding and dequantizing spectral components obtained from a transform or synthesis filter bank, and wherein said decoding and dequantizing steps employ fixed quantizer step sizes.

28. The method of claim 25, wherein a filter order and intervals of filter adaptation of said adaptive filter are selected suitable for irrelevancy reduction.

29. The method of claim 25, wherein the decoding stage for filter coefficients comprises a conversion from lattice coefficients or to Line Spectrum Pairs to linear-predictive coefficient filter coefficients.

30. An encoder for encoding a signal, comprising:

an adaptive filter controlled by a psychoacoustic model, said adaptive filter producing a filter output signal and having a magnitude response that approximates an inverse of the masking threshold; and

a quantizer/encoder for quantizing and encoding the filter output signal together with side information for filter adaptation control, wherein the spectral and temporal resolutions of one or more subbands utilized in said encoder are selected independent of said adaptive filter.

31. An encoder for encoding a signal, comprising:

an adaptive filter controlled by a psychoacoustic model, said adaptive filter producing a filter output signal and having a magnitude response that approximates an inverse of the masking threshold; and

a plurality of subbands suitable for redundancy reduction for transforming the filter output signal; and

a quantizer/encoder for quantizing and encoding the subband signals together with side information for filter adaptation control, wherein the spectral and temporal resolutions of one or more subbands utilized in said encoder are selected independent of said adaptive filter.

32. A decoder for decoding a signal, comprising:

a decoder/dequantizer for decoding and dequantizing said signal and decoding side information for filter adaptation control transmitted with said signal; and

an adaptive filter controlled by said decoded side information, said adaptive filter producing a filter output signal and having a magnitude response that approximates the masking threshold, wherein the spectral and temporal resolutions of one or more subbands utilized in said decoder are selected independent of said adaptive filter.

33. A decoder for decoding a signal transmitted using a plurality of subband signals, comprising:

a decoder/dequantizer for decoding and dequantizing said transmitted subband signals and decoding side information for filter adaptation control transmitted with said signal;

means for transforming said subbands to a filter input signal; and

an adaptive filter controlled by said decoded side information, said adaptive filter producing a filter output signal and having a magnitude response that approximates the masking threshold, wherein the spectral and temporal resolutions of one or more subbands utilized in said decoder are selected independent of said adaptive filter.

EVIDENCE APPENDIX

There is no evidence submitted pursuant to § 1.130, 1.131, or 1.132 or entered by the Examiner and relied upon by appellant.

RELATED PROCEEDINGS APPENDIX

There are no known decisions rendered by a court or the Board in any proceeding identified pursuant to paragraph (c)(1)(ii) of 37 CFR 41.37.

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SEP 21 2005
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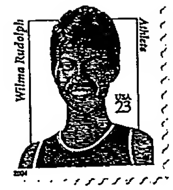

Ryan, Mason & Lewis, LLP
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FAIRFIELD, CT 06824

COPY

Receipt in the USPTO is hereby acknowledged of:

Transmittal Letter – (Original & 1 copy)
Notice of Appeal - (Original & 1 copy)
Petition for Extension of Time – (Original & 1 copy)

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JUL 21 2005

OIPE JC48
JUL 18 2005
PATENT & TRADEMARK OFFICE

Case Name: Edler 1-4
Serial No.: 09/586,072

1150-410

July 13, 2005 KMM

PTO/SB/31 (02-01)
Approved for use through 10/31/2002. OMB 0651-0031
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

NOTICE OF APPEAL FROM THE EXAMINER TO THE BOARD OF PATENT APPEALS AND INTERFERENCES		Docket Number (Optional) Edler 1-4	
I hereby certify that this correspondence is being deposited with the United States Postal Service with sufficient postage as first class mail in an envelope addressed to "Assistant Commissioner for Patents, Washington D.C. 20231" on July 13, 2005. Signature [signature] Typed or printed name Susan Fortuna	In re Application of Edler et al.		
	Application Number 09/586,072	Filed June 2, 2000	
	For ~~Perceptual Coding of Audio Signals Using Separated~~ Irrelevancy Reduction and Redundancy Reduction		
	Group Art Unit 2654	Examiner Qi Han	

Applicant hereby **appeals** to the Board of Patent Appeals and Interferences from the last decision of the examiner.

The fee for this Notice of Appeal is (37 CFR 1.17(b)) $500.00.

☐ Applicant claims small entity status. See 37 CFR 1.27. Therefore, the fee shown above is reduced by half, and the resulting fee is: $________.

☐ A check in the amount of the fee is enclosed.

☐ Payment by credit card. Form PTO-2038 is attached.

☐ The Commissioner has already been authorized to charge fees in this application to a Deposit Account. I have enclosed a duplicate copy of this sheet.

☑ The Commissioner is hereby authorized to charge any fees which may be required, or credit any overpayment to Deposit Account No. 50-0762. I have enclosed a duplicate copy of this sheet.

☑ A petition for an extension of time under 37 CFR 1.136(a) (PTO/SB/22) is enclosed.

WARNING: Information on this form may become public. Credit card information should not be included on this form. Provide credit card information and authorization on PTO-2038.

I am the

☐ applicant/inventor.

☐ assignee of record of the entire interest.
See 37 CFR 3.71. Statement under 37 CFR 3.73(b) is enclosed. (Form PTO/SB/96)

☑ attorney or agent of record.

☐ attorney or agent acting under 37 CFR 1.34(a).
Registration number if acting under 37 CFR 1.34(a).________.

[signature]
Signature

Kevin M. Mason
Typed or printed name

July 13, 2005
Date

NOTE: Signatures of all the inventors or assignees of record of the entire interest or their representative(s) are required. Submit multiple forms if more than one signature is required, see below*.

☐ *Total of ______ forms are submitted.